

Brekeke PBX

Version 2

Administrator's Guide (Advanced)

Brekeke Software, Inc.

Version

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1.	INTRODUCTION	6
2.	PROCESSES ON BREKEKE PBX	6
2.1.	Main Process.....	6
2.2.	Media Server Process	6
3.	ABOUT RTP RELAY	7
4.	SIP SERVER RELATIONSHIPS	8
4.1.	Brekeke PBX and SIP Server Relationship	8
4.2.	Default Dial Plans in the Bundled SIP Server	8
5.	CALL PROCESS FLOW	11
6.	NUMBER SYSTEM	13
6.1.	Defining a Brekeke PBX User	13
6.2.	Understanding Prefixes	13
7.	CALL FORWARDING	14
7.1.	Call Forwarding.....	14
7.1.1.	Example 1: Forwarding Calls to a Single Extension	14
7.1.2.	Example 2: Forwarding Calls to Multiple Extensions (Ring Group).....	14
7.1.3.	Example 3: Forwarding Calls to Multiple Group Extensions (Multiple - Ring Group).....	15
7.1.4.	Example 4: Enter SIP URI in the Forwarding Calls Field	15
7.1.5.	Example 5: Ignore Destination Number's Call Forward Settings-Forward Call Only to the Specified Extension Here.....	15
7.1.6.	Example 6: Ignore	16
7.1.7.	Example 7: Disconnection	16
7.2.	Call Forwarding (No answer)/ Call Forwarding (Busy)	16
7.2.1.	Example 1: Forwarding Unanswered/Busy Calls to Voicemail	17
7.2.2.	Example 2: Forwarding Unanswered/Busy Calls to Another Extension	17
7.2.3.	Example 3: Busy Call Forwarding is empty	17
7.3.	More Advanced Forwarding Settings	18
7.3.1.	Example 1: Ringer Time & Forwarding Destination (No answer/busy)	18
7.3.2.	Example 2: Ring Group and Call Forwarding	19
7.4.	Round Robin / Top-down Call Distribution	21

7.4.1.	Example 1: Round Robin / Top-Down	22
7.4.2.	Example 2: Round Robin / Top-Down	22
7.4.3.	Example 3: Call Queue	23
7.5.	Schedule Call Forwarding	23
7.5.1.	Schedule Setup	23
7.5.2.	Example DD (Date of months)	24
7.5.3.	Example: DD-DD (Date-Duration)	24
7.5.4.	Example: MM/DD (Specific Date).....	24
7.5.5.	Example: MM/DD-MM/DD (Specific Date- Period).....	24
7.5.6.	Example: Time.....	24
7.5.7.	Example: Caller ID information.....	25
7.6.	Add/Remove Forwarding Destinations (IVR).....	25
7.6.1.	Setting Up Add/Remove Forwarding Destinations.....	25
8.	SESSION MANAGEMENT	26
8.1.	Unlimited Sessions.....	26
8.2.	Limited Sessions (0-6).....	26
8.3.	Using Maximum Session Setting	26
9.	CALL HOLD / CALL TRANSFER.....	27
9.1.	Different Methods of Call Hold / Call Transfer	27
9.1.1.	Call Hold by re-INVITE/Call Transfer by REFER.....	27
9.1.2.	Call Hold/ Call Transfer using Keypad Commands	27
9.2.	Call Hold	27
9.2.1.	Call Hold – Activation / Deactivation.....	27
9.3.	Initiating Call Transfer	28
9.3.1.	Initiating Call Transfer 1 – Calling the Third Party.....	28
9.3.2.	Initiating Call Transfer 2 – Calling Multiple Users	28
9.4.	Processing Call Transfer	29
9.4.1.	Processing Call Transfer 1 – Attended Transfer	29
9.4.2.	Processing Call Transfer 2 – Unattended Transfer.....	29
9.5.	Canceling Call Transfer.....	30
9.5.1.	Canceling Call Transfer 1 – Canceling Transfer While Talking with a 3rd Party	30
9.5.2.	Canceling Call Transfer 2 – Cancel Transfer While Ringing the 3rd party.....	30
9.6.	Cautions for Call Transfer.....	31

9.6.1.	Extending Call Transfer	31
10.	AUTO ATTENDANT	32
10.1.	Setting Up Auto Attendant	32
10.1.1.	Accessing Auto Attendant	32
10.2.	Cautions for Auto Attendant.....	32
10.2.1.	Call Transfer to Auto Attendant	32
10.2.2.	Call Transfer from Auto Attendant to the Non-Brekeke PBX User.....	33
10.3.	Speed Dial Setting Example for Auto Attendant.....	33
10.3.1.	Simple Speed Dial settings	33
10.3.2.	Speed Dial settings using Regular Expressions	34
11.	VOICEMAIL	35
11.1.	Voicemail Inbox Number	35
11.2.	Accessing Voicemail	35
11.3.	Greeting Messages.....	35
11.3.1.	Recording / Uploading Greeting Messages	35
11.3.2.	Selecting a Greeting Message	35
11.4.	Managing Messages.....	36
11.5.	Retrieving Voicemail Messages	36
11.6.	Voicemail Message Status	36
11.6.1.	New Message Status.....	36
11.6.2.	Saved Message Status	36
11.7.	Email Notification	37
11.7.1.	Set Up Email Notification	37
11.7.2.	Send Email Notification of the Messages from Call Recording	37
11.8.	Voicemail Messages: Forwarding and Email Notification	37
11.8.1.	Forwarding Voicemail Message.....	37
11.8.2.	Email Notifications.....	37
11.8.3.	Max Messages	38
11.9.	Voicemail Inbox Access Limitation	38
11.10.	Creating / Deleting Voicemail Inboxes	38
12.	AUTOMATIC ROUTE SELECTION (ARS).....	39
12.1.	Route Search Process	40
12.2.	Matching Patterns.....	41

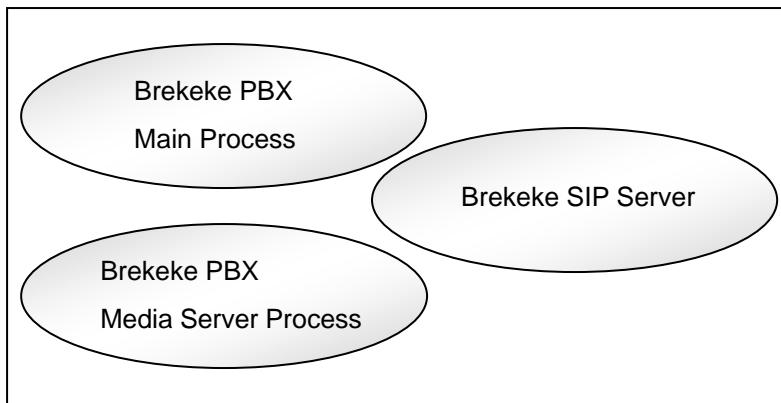
12.3.	Deploy Patterns	41
12.4.	ARS Templates and Editing Variables	42
13.	THIRD PARTY CALL CONTROL	43

1. Introduction

This document is for system administrators who have already reviewed the Brekeke PBX Administrator's Guide (Basic). This document provides more detailed and advanced information about each function of the Brekeke PBX.

2. Processes on Brekeke PBX

When Brekeke PBX is started, three processes are initiated; SIP Server process, Brekeke PBX main process, and the process for media server (voicemail/IVR features).



2.1. Main Process

The Brekeke PBX main process receives and dispatches SIP sessions between SIP User Agents (UAs). All call sessions between caller and callee are mediated by the Brekeke PBX main process.

2.2. Media Server Process

The media server process handles some Brekeke PBX features, such as voicemail, and auto attendant.

3. About RTP Relay

Depending on your operating environment and its requirements, and to ensure proper SIP communications, the RTP relay through Brekeke PBX may be necessary.

- ◆ If you are using a SIP UA that does not support the SIP-INFO (DTMF-relay) method, and wish to use Brekeke PBX features using key pad commands, such as Call Forwarding, Call Park, Call Recording, etc., RTP packets need to be relayed through the Media Server.
- ◆ If you are using SIP devices (SIP UA, SIP proxy server, etc.) that do not support changing RTP sender information by receiving re-INVITE requests, RTP packets need to be relayed through the Media Server.
- ◆ If you are using SIP devices that do not allow changing a voice codec during a call, select “on (G.711u only)” at RTP relay setting or “0 (G.711u only)” at Coded priority in Brekeke PBX setting. Selecting “off” at RTP relay setting may not work well depending on SIP devices and there may be compatibility issues between different types of SIP devices. Thorough testing is recommended before deciding to set RTP relay setting to “off” (which is not G.711u only) options or use a codec other than G.711u.

RTP relay settings can be set under several locations in Brekeke PBX: Options, ARS Settings, and User Settings. The settings under [User Settings] have the highest priority. When the default is set at [User Setting], settings under [ARS Settings] are applied. When User Settings and ARS Settings are set to the defaults, the settings for [Options] are applied.

RTP relay will be turned off only when both end points are set to RTP relay “off”. When one of the end points require RTP relay, RTP will be relayed through Brekeke PBX. By decreasing the amount of RTP packets handled by Brekeke PBX, the Brekeke PBX machine load will increase. Even when the RTP relay setting is set to “off”, Brekeke PBX handles RTP for some features such as Music on Hold and Voicemail. Thus, depending upon the type of usage and environment, the maximum number of concurrent session can vary for each setting.

Brekeke PBX supports not only codec G.711 u-law but also G.729 (Optional), G.711 A-law and iLBC. When the RTP relay setting is set to “on”, Brekeke PBX will do codec conversions between different codecs. So, both parties can talk each other without problems even when they are using different codecs.

4. SIP Server Relationships

4.1. Brekeke PBX and SIP Server Relationship

A SIP server is embedded in Brekeke PBX. If you log in to Brekeke PBX as an Administrator, you can also access and manage the bundled SIP server. The bundled SIP server is a SIP proxy server which provides call session management in a VoIP network and processes SIP requests and responses. When Brekeke PBX receives a call session, it initiates another session with the appropriate destination according to user settings, such as forwarding destinations, then joins the two sessions.

4.2. Default Dial Plans in the Bundled SIP Server

The bundled SIP server has the following default settings in the Dial Plan.

PBX Prefix

If the session that has come from Brekeke PBX has the prefix “pbx” in the “To” header, the prefix will be stripped off and the session will be routed to Brekeke PBX again. Authentication for this session is disabled.

From PBX Page (Versions 2.2 or later)

If the session from Brekeke PBX has a “page” parameter, then the PBX will set related parameters to trigger the auto answer function on the phone.

From PBX REG (Between version 2.2.* and 2.2.7.8)

If the INVITE request is from the Brekeke PBX and the callee is registered, then B2BUA mode will be set to false and Authentication will be disabled.

From PBX

This dial plan rule will be applied to the sessions coming from Brekeke PBX that do not match the Dial Plan rules above it. It will disable the Authentication.

From Brekeke PBX v2.2, all sessions through “From PBX” will set B2BUA mode to true by default. From Brekeke PBX v2.2.7.8, B2BUA mode is the “not” operation of callee registration status.

To PBX From ITSP (Versions 2.2.7.8 or later)

All sessions from registered ITSP or SIP devices will be routed to Brekeke PBX. B2BUA mode will be set to true and Authentication to false.

To PBX REG (Between version 2.2.* and 2.2.7.8)

If the INVITE request is not from Brekeke PBX and the caller is a registered client, then the call will be routed to Brekeke PBX and B2BUA mode will be set to false.

To PBX

The sessions that are not from Brekeke PBX will be routed to Brekeke PBX with UDP transport. The "To" header field will not be changed. From Brekeke PBX v2.2, all sessions through "To PBX" will set B2BUA mode to true by default. From Brekeke PBX v2.2.7.8, the "To PBX" rule will be applied to sessions from registered callers and B2BUA mode and Authentication will be set to false.

To PBX auth (Versions 2.2.7.8 or later)

This dial plan rule will be applied to the sessions going to Brekeke PBX that do not match the Dial Plan rules above it. It will enable Authentication.

Subscribe

SUBSCRIBE requests for Message Waiting Indicator (MWI), presence, BLF, SCA and for Multiline functions of SAXA IP Phones will be routed to Brekeke PBX.

Thru Register

REGISTER requests from Brekeke PBX will be applied when Brekeke PBX is registering to the third party SIP proxy server using Thru Registration. When the third party SIP proxy server has a global IP address, and if the bundled SIP server does not have interface for the global IP address, the bundled SIP server does not send REGISTER requests to the third party SIP proxy server. This is to prevent problems with ITSPs since having a local IP address registered at an ITSP's proxy server tends to cause unnecessary problems.

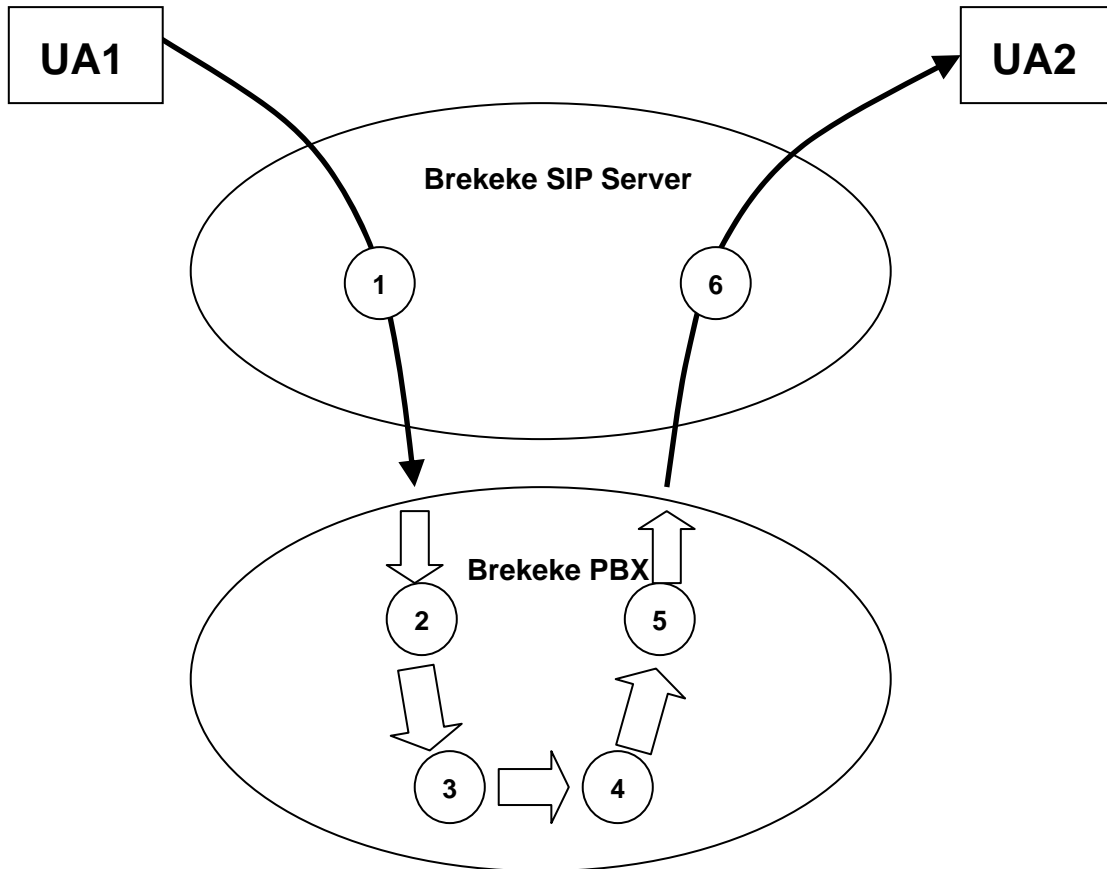
Register for PAL (Versions 2.2 or later)

If Brekeke PBX license has Brekeke PAL option, registering continues without changing anything.

- ✓ *Dial Plan called “To MediaServer” is not required from Brekeke PBX v2.0. When the prefix for media server is set at Brekeke PBX, it directly connects with the media server without having SIP proxy in between.*
- ✓ *From Brekeke PBX v2.1, the Dial Plan called “From PBX 1” is not necessary. “From PBX 2” in v2.0 is renamed as “From PBX” from v2.1.*
- ✓ *From Brekeke PBX v2.3, Dial Plans called “From PBX REG” and “To PBX REG” are not needed.*

5. Call Process Flow

In this section, we explain how a call from UA1 to UA2 is processed. This graphic image represents the user agent's call step-by-step.



The following items correlate to the image presented above:

- 1) The call initiated by UA1 matches with the Dial Plan rule "To PBX (***)" at the bundled SIP server and is routed to the Brekeke PBX.
- 2) When Brekeke PBX receives the call, it searches for an ARS "Pattern IN" that matches with the call. If Brekeke PBX finds a pattern, it will change the caller's address (From) and call recipient's address (To) as specified in the Deploy Patterns.
- 3) If the "To" header includes a prefix for Call Pickup, Park Pickup, or for accessing an existing call, Brekeke PBX will make the call join the corresponding existing call. If not, Brekeke PBX will treat the call as a new call.

- 4) Brekeke PBX will check the settings in [Users > Edit] > [Forwarding destinations*] or [Forwarding destination (No answer/Busy)] and determine the call destination. If multiple destinations are set in [Forwarding destinations*] or some extension is set in [Automatic Monitoring*] field, Brekeke PBX will process the call routings for multiple destinations in parallel with Step 5.
- 5) Brekeke PBX searches for an ARS "Pattern OUT" that matches the call. If Brekeke PBX finds a pattern, it will change the caller's address (From) and call recipient's address (To) as specified in the Deploy Patterns.
- 6) The call is processed by the bundled SIP server's Dial Plan feature and routed to UA2.

6. Number System

6.1. Defining a Brekeke PBX User

Brekeke PBX recognizes each extension number as a Brekeke PBX user. A Brekeke PBX user is the same as the user-info part of the SIP URI (e.g., In the SIP URI “sip:001@brekeke.com”, 001 is considered as a Brekeke PBX user). Brekeke PBX users can use Brekeke PBX features, such as Call Transfer, Call Hold, Call Forwarding, and Voicemail.

6.2. Understanding Prefixes

To provide easy access to Brekeke PBX features such as Voicemail and Auto Attendant, Brekeke PBX uses a prefix system. By adding the appropriate prefix to an extension number, users can directly access Voicemail or Auto Attendant.

The table below shows the default prefix settings:

IVR prefix	ivr
Voicemail prefix	vm
Voicemail review / Setting prefix	msg
Call Pickup prefix	*

If you upgraded the Brekeke PBX from version 1.x to 2.x and kept all configurations, the prefixes are the same as before.

IVR prefix	6
Voicemail prefix	7
Voicemail review / Setting prefix	8
Call Pickup prefix	*

Though it is possible to change prefix values in the [Options] menu, we do not recommend changing the values unless necessary for compatibility with older versions.

From version 2.0, the prefixes are not numbers. So it is not possible to dial the prefix directly from most types of SIP devices. We have prepared the default ARS rule “mediaserver_prefix” so that users can dial numbers to directly access the Media Server features (Voicemail/IVR).

7. Call Forwarding

Call Forwarding also includes Call Forwarding (No Answer) and Call Forwarding (Busy).

7.1. Call Forwarding

[Administrative settings] > [Type of Call Forwarding] = Basic

Each user can set up call forwarding individually. When a call is received at the user's extension, the call will be forwarded to the destination set in the **[Forwarding destinations*]** field in the **[Users > Edit]** page.

By default, the **[Forwarding destinations*]** field is left blank and all incoming calls are directed to the user's extension. If there are destinations specified in the field, only those extensions will receive calls.

If multiple destinations are set, incoming calls will be forwarded to all destinations (Ring Group). Each extension's phone will ring at the same time, and the first person to answer will be connected with the caller.

7.1.1. Example 1: Forwarding Calls to a Single Extension

Extension: 001

Forwarding destinations*	002
---------------------------------	-----

With this setting, every call that comes into extension 001 is transferred to extension 002. Extension 001 will not receive any calls that are dialed for extension 001.

7.1.2. Example 2: Forwarding Calls to Multiple Extensions (Ring Group)

Extension: 001

Forwarding destinations*	001, 002
---------------------------------	----------

With this setting, every call that comes into extension 001 will be transferred to extensions 001 and 002. If there is no forwarding set at the extension 002, both extension 001 and 002 will receive calls that comes into extension 001. The first person that answers the phone will be connected with the caller.

7.1.3. Example 3: Forwarding Calls to Multiple Group Extensions (Multiple - Ring Group)

Extension: 001

Forwarding destinations*	100, 200
---------------------------------	----------

Extension: 100

Forwarding destinations*	101, 102, 103
---------------------------------	---------------

Extension: 200

Forwarding destinations*	201, 202, 203
---------------------------------	---------------

With this setting, incoming calls to extension 001 are forwarded to extensions 100 and 200. Extensions 100 and 200 in turn forward calls to extensions 101,102, 103, 201, 202, and 203. If the users on these extensions do not have call forwarding set up, these 6 extensions will ring. The first person that picks up the phone will be connected with the caller.

7.1.4. Example 4: Enter SIP URI in the Forwarding Calls Field

Extension: 001

Forwarding destinations*	sip:6636@brekeke.com
---------------------------------	----------------------

You can specify SIP URI in the [Forwarding destinations*] field under the [users > Edit] page to forward the call to another SIP service.

7.1.5. Example 5: Ignore Destination Number's Call Forward Settings-Forward Call Only to the Specified Extension Here

Extension: 001

Forwarding destinations*	%002
---------------------------------	------

With this setting, the calls directed to extension 001 are forwarded to extension 002. All of the call forwarding settings, [Ringer Time] and [Keypad Commands] at extension 002 will be ignored, and the settings at 001 will be applied. Also, forwarded calls will be recorded on extension 001's call log.

7.1.6. Example 6: Ignore

Extension: 001

Forwarding destinations*	001, !002
---------------------------------	-----------

Adding “!” before an extension instructs the system to ignore that extension. This setting is useful when a certain extension needs to be excluded from the [Forwarding destinations*] setting temporarily. Note that the call will be forwarded to the destination set for Forwarding destination (No answer) when all of the destination extensions have “!”

7.1.7. Example 7: Disconnection

Extension: 001

Forwarding destinations*	!
Forwarding destination (No answer)	

If only ! is set in the [Forwarding destinations*] field, [Forwarding destination (No Answer)] setting for the extension 001 will be applied. If none is set in the [Forwarding destination (No Answer)] field, the call will be disconnected.

7.2. Call Forwarding (No answer)/ Call Forwarding (Busy)

These forward settings are used in the following situations:

- ◆ No answer after the ringer timeout occurs --> Call Forwarding (No answer)
- ◆ Calls can not be established or received busy response --> Call Forwarding (Busy)

Each user can set up forwarding for unanswered/busy calls. By default, unanswered/busy calls are forwarded to the user’s voicemail inbox. (For the extension 001, “vm001” is set for the [Forwarding destination (No answer)] and [Forwarding destination (Busy)] field.

Only a single destination can be assigned in the [Forwarding destination (No answer)] and [Forwarding destination (Busy)] field. If you need to set up multiple destinations to be used for these features, please refer to the method described in section 7.1.3 “Example 3: Forwarding Calls to Multiple Group Extensions (Multiple - Ring Group)”.

7.2.1. Example 1: Forwarding Unanswered/Busy Calls to Voicemail

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Incoming calls to extension 001 are forwarded to the user's voicemail inbox (extension vm001) if the call is busy or remains unanswered after ringing for 15 seconds.

7.2.2. Example 2: Forwarding Unanswered/Busy Calls to Another Extension

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	916504016636
Forwarding destination (Busy)	vm001

With this setting, the call will be forwarded to 916504016636 if the call remains unanswered for 15 seconds. If the call is busy, the call will be forwarded to the extension 001's voicemail.

7.2.3. Example 3: Busy Call Forwarding is empty

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	

Incoming calls to extension 001 are forwarded to the user's voicemail inbox (vm001) if the call remains unanswered after ringing for 15 seconds. When the field for [Forwarding destination (Busy)] is left blank, even for the case when the call is busy, Brekeke PBX will wait for 15 seconds and then the call will be forwarded to the user's voicemail.

7.3. More Advanced Forwarding Settings

By combining Call Forwarding, Call Forwarding (No answer), Call Forwarding (Busy) and Ringer time, you may create more advanced call forwarding settings.

Here are a few rules that are useful to remember:

- ◆ When Call Forwarding is set for the extension, the Ringer time for the call depends on the setting on the extension that calls are forwarded to. If no one answers the call for the Ringer time, original call recipient (not forwarding destination)'s [Forwarding destination (No answer)] will be applied (Please refer to section 7.3.1.) If all of the forwarding destinations could not establish a call or if all forwarding destinations are busy, [Forwarding destination (Busy)] will be applied.
- ◆ Only a single destination can be set in the [Forwarding destination (no answer/busy)] field.
- ◆ After the call is forwarded by Busy Call Forwarding or No Answer Call Forwarding, forwarding destination's [Forwarding destinations], [Forwarding destination (No answer)], [Forwarding destination (Busy)] settings will be used and the call can be transferred to another destination. (Please refer to section 7.3.2).

7.3.1. Example 1: Ringer Time & Forwarding Destination (No answer/busy)

Extension: 001

Forwarding destinations*	101, 102
Ringer time (sec)	20
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Extension: 101

Forwarding destinations*	
Ringer time (sec)	15
Forwarding destination (No answer)	vm101
Forwarding destination (Busy)	vm101

Extension: 102

Forwarding destinations*	
Ringer time (sec)	25
Forwarding destination (No answer)	vm102
Forwarding destination (Busy)	vm102

- 1) All incoming calls to the extension 001 > forwarded to both extensions 101 and 102
- 2) Extension 101 rings for 15 seconds and Extension 102 rings for 20 seconds > no answer
- 3) The call will be forwarded to the 001's voicemail inbox vm001

User 001's settings, Ringer time and Call Forwarding (No answer), are treated as highest priority even when the user 101 and 102 have set up call forwarding (No answer) on all of his/her calls. Ringer time for extension 001 is set for 20 seconds. Even after User 101's ringer stops, extension 102 continues ringing to 20 seconds. Then the call will be forwarded to extension 001's voicemail inbox (extension vm001).

With the settings above, incoming calls to extension 001 will be transferred to both extensions 101 and 102. If the call is not answered by extension 101 or 102, it will be forwarded to the destination specified in User 001's [Call Forwarding (No answer)] field. [Call Forwarding (No answer)] settings for extensions 101 and 102 will not be used.

7.3.2. Example 2: Ring Group and Call Forwarding

Extension: 000

Forwarding destinations*	001
Ringer time (sec)	5
Forwarding destination (No answer)	001A
Forwarding destination (Busy)	001A

Extension: 001A

Forwarding destinations*	100A, 200A
Ringer time (sec)	10
Forwarding destination (No answer)	001B
Forwarding destination (Busy)	001B

Extension: 001B

Forwarding destinations*	100B, 200B
Ringer time (sec)	10
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Extension: 100A

Forwarding destinations*	101, 102, 103
---------------------------------	---------------

Extension: 200A

Forwarding destinations*	201, 202
---------------------------------	----------

Extension: 100B

Forwarding destinations*	111, 112, 113, 114, 115, 116
---------------------------------	------------------------------

Extension: 200B

Forwarding destinations*	211, 212, 213, 214
---------------------------------	--------------------

- 1) All incoming calls to extension 000 are forwarded to extension 001.
- 2) When a call is unanswered for 5 seconds, it is forwarded to 001A.
- 3) The call is forwarded to 100A and 200A.
- 4) Extensions 101, 102, 103, 201, and 202 will ring.
- 5) If extension 001's [Ringer Time] is more than 5 seconds, 001 will continue to ring even after 5 seconds.
- 6) If the call is unanswered for 10 seconds it is forwarded to extension 001B.
- 7) The call is forwarded to 100B and 200B.
- 8) Extensions 111, 112, 113, 114, 115, 116, 211, 212, 213, and 214 will ring.
- 9) If the call is unanswered for 10 seconds, it is forwarded to the extension 001's voicemail inbox vm001.
- 10) When vm001 answers the call, all ringing will stop.

Meanwhile all extensions that receive the call will keep ringing until their own Ringer time setting expires or the call is answered.

- ✓ *With this example, we used letters for the extension numbers in order to clarify different groups.*
- ✓ *If a caller is using a SIP UA which allows letters, the caller can make a call directly to a number such as 100B.*

7.4. Round Robin / Top-down Call Distribution

Brekeke PBX features two types of call distribution: Round Robin and Top-Down. The round robin setting is useful if you want to distribute calls equally within a specific group of extensions. The top-down setting is useful if you want to distribute calls in a certain order within the specific group of extensions. In either case, the system will ring each available extension sequentially until the call is answered.

Round Robin

Under round robin settings, the extension that rings first will keep changing in the order as specified. Every time the call will be forwarded to the next specified extension of the one that answered the last call.

Top-Down

Under top-down setting, calls are always forwarded in a specified order. With Brekeke PBX, the number specified in the left most in the field is given the highest priority. The same setting can be configured under the basic call forward settings, but top-down settings are more simplified.

Single Attempt

When the Single Attempt setting is set to “no” (default setting), calls will keep ringing the next destination even after the [Ringer time] has expired. The same effect can be achieved by specifying multiple destinations under “Basic” settings located in the [Type of Call Forwarding] field; however, using this feature is more straightforward.

When the Single Attempt setting is set to “yes”, after one failure to establish a connection, the calls will be forwarded to the destination set at [Forwarding destination (No answer)] or [Forwarding destination (Busy)].

Call Queue

Using the Call Queue feature, calls are kept on hold until a specified user answers the call. If the Single Attempt setting is set to “yes”, the unanswered call will be forwarded to the destination set at [Forwarding destination (No answer)] after one attempt.

7.4.1. Example 1: Round Robin / Top-Down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Busy/No Answer Forwarding	vm100
Waiting time in the queue (sec)	0
Single attempt	no

Top-Down Setting

- 1) Extension 100 starts ringing.
- 2) If the call is unanswered at extension 100 after 10 seconds, extension 101 starts ringing.
- 3) If the call is unanswered at extension 101 after 5 seconds, extension 102 starts ringing.
- 4) If the call is unanswered at extension 102 after 5 seconds, the call is forwarded to vm100.

Round Robin Setting

The Call Forwarding order is the same as the top-down setting for the first call, however, from the second call on, the call is forwarded to the extension after the one that answered the most recent call.

7.4.2. Example 2: Round Robin / Top-Down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Busy/No Answer Forwarding	vm100
Waiting time in the queue (sec)	0

Extension: 101 (Call Forwarding setting [Basic])

Forwarding destinations*	200,201
Ringer times (sec)*	20

Call Forwarding settings are enabled at each extension individually, regardless of the settings of the extension the call is forwarded from. With the above example, when a call is forwarded to extension 101, it is directed to extension 200 and 201 simultaneously. If the call is unanswered at both extensions after 5 seconds, extension 102 will ring. Extension 200 and 201 will keep ringing until they reach the time specified in the Ringer times.

7.4.3. Example 3: Call Queue

Extension 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	20
Busy/No Answer Forwarding	vm100
Waiting time in the queue (sec)	180
Max number of calls in the queue	10

The Call Queue feature is used when the time, more than 1 second, is set at the [Waiting time in the queue (sec)] field.

Calls are answered by the available extensions among 100, 101, or 102. The order the extensions answer will depend upon the mode setting (Round Robin or Top-Down). If the call is unanswered, it is placed in the queue.

When the calls in the queue exceed 10, the 11th call is forwarded to vm100. The callers in the queue will hear hold music while waiting to be answered. If the call is not answered after the caller waits 180 seconds in the queue, it will be forwarded to vm100.

After the period set in the [Call interval (msec)], the call in the top of the queue will be forwarded to the available extension. When [Single attempt] setting is set for "no", the call will keep calling until the waiting time for the queue (180 seconds) expires. When [Single attempt] setting is set for "yes", the call will be forwarded to the destination specified under [Busy/No Answer Forwarding] after one failed attempt.

7.5. Schedule Call Forwarding

When Schedule is selected for the [Type of Call Forwarding], call forwarding rules can be defined using conditions based on date, time, and Caller ID information.

7.5.1. Schedule Setup

Schedule setting is applied when the specified condition (period, date, hour, or Caller ID) occurs. (The action will be same as the basic call forwarding setting.)

To set up a schedule:

- 1) Select *schedule* as the [Type of call forwarding] in the [Users > Edit] page.
- 2) Click the [Save] button, and menu items [Schedule Settings 1] and [Schedule Setting 2] will appear.

- 3) To define a schedule, click the [...] button. This will open the schedule menu.
- 4) Choose the dates and times during which you would like calls to be forwarded. When set [Days of week], specify which day of which week in a month. For example: Checking First and Sunday means the schedule will only apply to the first Sunday in every month.
- 5) Set call forwarding destinations as you would with the other call forwarding settings.

7.5.2. Example DD (Date of months)

5,10,15,20,25

This setting will be applied on the 5th, 10th, 15th, 20th, and 25th of every month during the specified term.

7.5.3. Example: DD-DD (Date-Duration)

5-10,15-20

This setting will be applied on between the 5th and 10th; and the 15th and 20th of every month during the specified term.

7.5.4. Example: MM/DD (Specific Date)

5/10,5/20,6/10,6/20

This setting will be applied on May 10th, May 20th, June 10th, and June 20th.

7.5.5. Example: MM/DD-MM/DD (Specific Date- Period)

5/10-6/20,8/10-8/20

The setting will be applied between May 10th and June 20th, and between August 10th and August 20th during the specified term.

7.5.6. Example: Time

08:30 – 13:30

23:30 – 00:30

This setting will be applied between 8:30am and 1:30pm; and between 11:30pm and 12:30am.

7.5.7. Example: Caller ID information

Applies to (Caller numbers)*	3*
Not Applies to (Caller numbers)*	3001,3002

By adding * (wildcard) after a number, you can specify all numbers that starts with that number.

By adding "?", you can specify any one number.

With the setting example above, all the coming calls that start with 3 except 3001 and 3002 will be forwarded according to this schedule call forwarding setting. The numbers specified in the "Not Applies to (Caller numbers)" field override "Applies to (Caller numbers)". If the "Applies to (Caller numbers)" field is left blank, all incoming calls will be directed according to this schedule call forwarding setting.

7.6. Add/Remove Forwarding Destinations (IVR)

With this feature, a user can add or remove their own number in [Forwarding destinations*] from their phone instead via Brekeke PBX Admintool page.

7.6.1. Setting Up Add/Remove Forwarding Destinations

Extension: 100

Forwarding destinations	%ivr100
IVR	Add/Remove forwarding destinations
Target users*	300,400

When extension 200 calls into extension 100, extension 200 is added to the forwarding destinations for extension 300 and 400. If extension 200 already exists in the forwarding destinations for those numbers, extension 200 will be invalid as a forwarding destination (remain as forwarding destination with "!" in the front of the extension number).

8. Session Management

You can set a maximum number of sessions that an extension can receive in [Max sessions] under the Brekeke PBX user settings. Call sessions are counted between the beginning of the conversation until the phone call is terminated, or when the phone begins ringing until the phone call is terminated.

8.1. Unlimited Sessions

When unlimited sessions are set at **[Max sessions]**, incoming calls will ring the extension even if the extension is in session(s). The call is forwarded to the destination set at **[Forwarding destination (No answer/Busy)]** when it is unanswered by the extension after the specified time (Ringer time) or conditions (SIP UA send response as “busy”, “reject”, etc.).When **[Type of Call Forwarding]** is set for “Round robin/Top-down”, the call will only ring the available extensions.

8.2. Limited Sessions (0-6)

If the callee’s extension already has the maximum number of sessions (set at [Max sessions]) and all sessions are busy, the call will be forwarded to the destination set at **[Forwarding destination (No answer/Busy)]**. The calls will stay in pending status if there is a session with ringing status until all sessions are occupied or there is no session remained to call.

8.3. Using Maximum Session Setting

This section covers when to use Maximum Session settings.

Preventing missed calls

By setting [Max sessions], users can avoid missing calls. When multiple calls are coming in at the same time, all calls, except the first call, are forwarded to the destination set at [Forwarding destination (Busy)].

Setting the preferred number of simultaneous calls

For the SIP UAs that have capacity to handle multiple sessions, users can limit the preferred number of simultaneous calls and make other lines available for outgoing sessions.

9. Call Hold / Call Transfer

9.1. Different Methods of Call Hold / Call Transfer

9.1.1. Call Hold by re-INVITE/Call Transfer by REFER

Many SIP devices support Call Hold by re-INVITE (HOLD button) and/or Call Transfer by REFER (TRANSFER button). You can use these features with Brekeke PBX.

When using Brekeke PBX, re-INVITE/REFER requests are processed at the Brekeke PBX. Thus, Call Hold and Call Transfer will work even if the other party's device does not support re-INVITE or REFER.

9.1.2. Call Hold/ Call Transfer using Keypad Commands

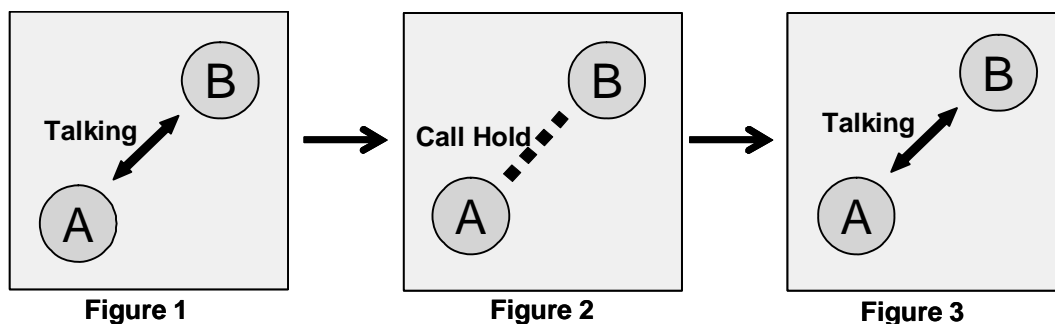
Even if SIP devices do not support the method of Call Hold by re-INVITE / Call Transfer by REFER, these devices can send commands to Brekeke PBX by using DTMF (Keypad) for Call Hold / Call Transfer. On some devices, you may be able to assign DTMF numbers to HOLD/TRANSFER buttons and use them for Brekeke PBX proprietary method of Call Hold / Call Transfer. Keypad commands for Brekeke PBX are listed in the Brekeke PBX Users Guide.

9.2. Call Hold

Any extension using the Call Hold feature with keypad commands must be registered as a user extension of Brekeke PBX. Call Hold permissions are set in the [Users >Edit] > **[Keypad Commands]** option. In this section, Call Hold is refers to the unique Call Hold feature of the Brekeke PBX, and it is different from the Call Hold using re-INVITE requests (Section 9.1).

9.2.1. Call Hold – Activation / Deactivation

When Brekeke PBX user extensions A and B are on the phone with each other (Figure 1), either user can place the call on hold by using Keypad Commands (Figure 2). Taking a call off hold can only be done by the user who placed the call on hold, so the conversation will not be resumed until User A takes User B off hold.



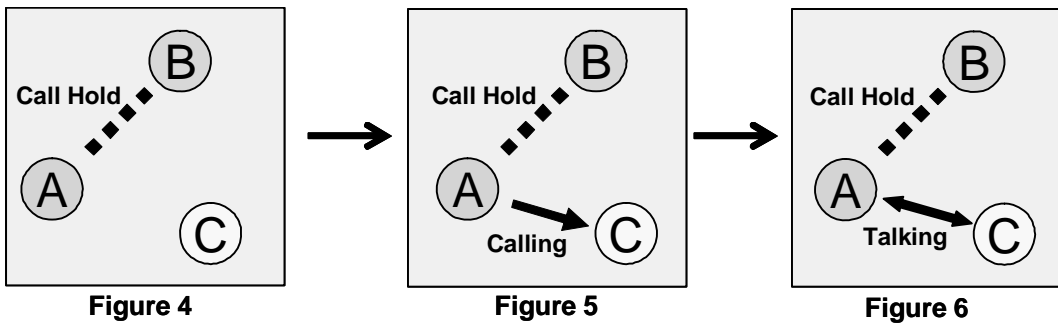
9.3. Initiating Call Transfer

After putting a call on hold, Brekeke PBX users can transfer a call to other users.

- ✓ *In this section, Call Hold is referring to the unique Call Hold feature of Brekeke PBX, and it is different from Call Hold using REFER requests (Section 9.1).*

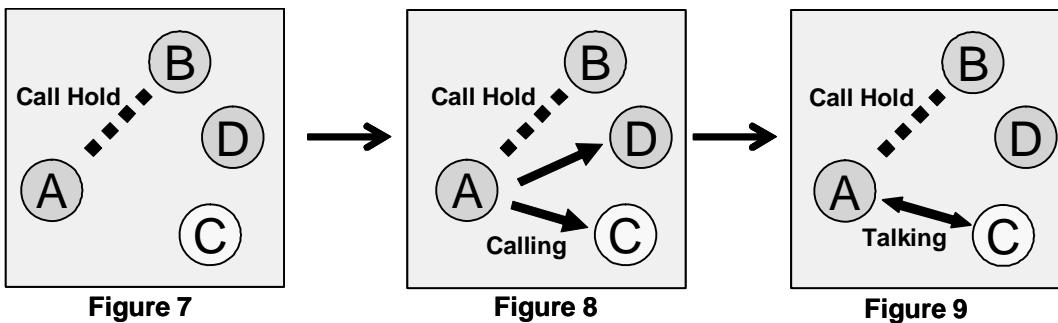
9.3.1. Initiating Call Transfer 1 – Calling the Third Party

User A puts the call with User B on hold (Figure 4), and initiates a new session with User C (Figure 5). When User C answers the call, the session between User A and User C will be connected (Figure 6).



9.3.2. Initiating Call Transfer 2 – Calling Multiple Users

If User C sets a [Forwarding destinations] for his/her incoming calls, the calls will be transferred to the specified destinations. In this example, User C has set User C and User D for the destinations. (Figure 7 and Figure 8) The first user who answers the phone will be connected with User A.



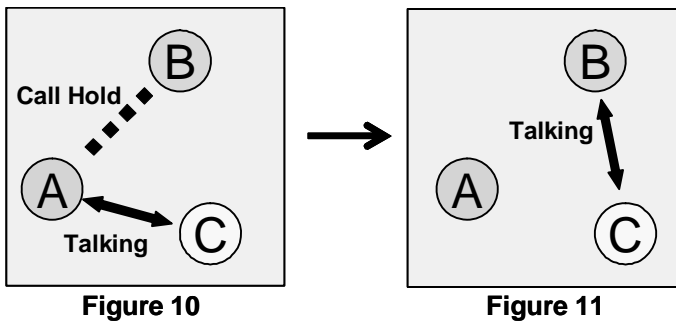
9.4. Processing Call Transfer

After talking with the transfer target user, you will have the opportunity to go back to the original session with User B.

- ✓ *In this section, Call Hold refers to the unique Call Hold feature of Brekeke PBX, and it is different from Call Hold using REFER requests (Section 9.1).*

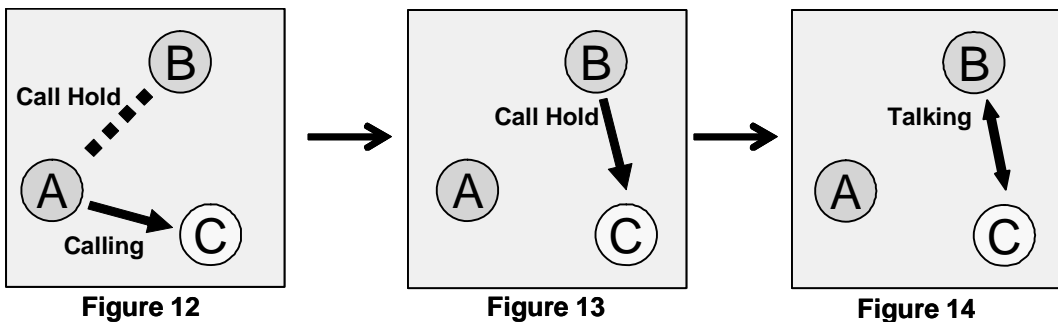
9.4.1. Processing Call Transfer 1 – Attended Transfer

When User A initiated Call Transfer, the call between Users A and B was put on hold. When User A hangs up the phone while Users A and B's session is connected (Figure 10, similar situation with Figure 6 and 9), the call with Users A and B will be transferred to User C. Now Users B and C are connected.



9.4.2. Processing Call Transfer 2 – Unattended Transfer

While the session between Users A and B are on hold, User A tries to connect with User C. (Figure 12, similar situation with Figure 6, 9, and 10) User A hangs up the phone before User C answers the call, then the call will be transferred to User C and User B wait for User C to answer. (Figure 13)



- ✓ *Using Unattended Transfer, when User C did not answer the phone, the call between User A and B will be reestablished (User A's phone will ring).*

9.5. Canceling Call Transfer

You may cancel Call Transfer in the middle of the process, and either go back to talk with the original caller or transfer the call to the different destination.

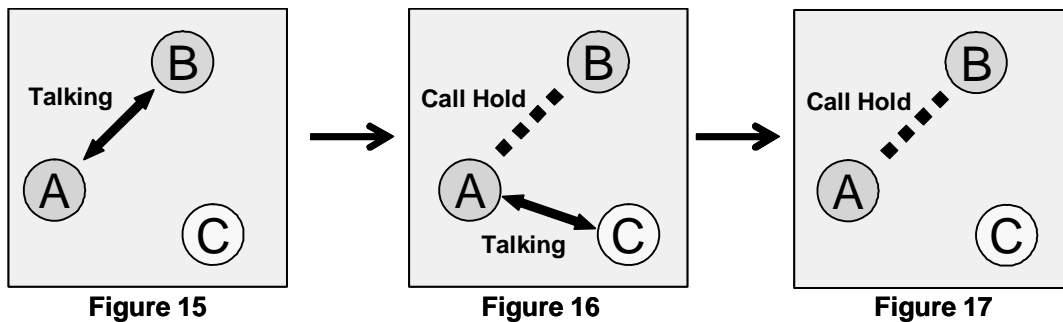
- ✓ *In this section, Call Hold refers to the unique Call Hold feature of Brekeke PBX and is different from Call Hold using REFER requests (Section 9.1).*

9.5.1. Canceling Call Transfer 1 – Canceling Transfer While Talking with a 3rd Party

There are two ways to cancel Call Transfer when you are talking with a 3rd party.

- ◆ User A executes the cancel Call Transfer command
- ◆ User C hangs up call with User A

When the Call Transfer process was canceled, while User B was on hold (Figure 16, similar situation with Figure 6, 9, and 10), the call will be put back to the hold state with User A and B. (Figure 17)

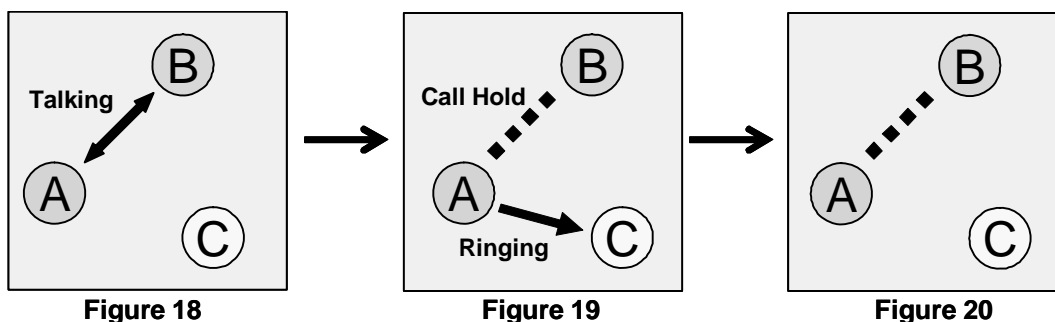


9.5.2. Canceling Call Transfer 2 – Cancel Transfer While Ringing the 3rd party

There are two ways to cancel Call Transfer when session is ringing for the 3rd party

- ◆ User A executes the cancel Call Transfer command
- ◆ When a call to User C cannot be completed (User C's phone is not connected, Call blocking, etc.)

You can cancel Call Transfer while the phone is ringing at the destination, User C (Figure 19, similar situation at Figure 5 and 8). The call status goes back to the hold status between Users A and B.



9.6. Cautions for Call Transfer

9.6.1. Extending Call Transfer

In the state of Figure 21 with regular settings, user C cannot initiate a Call Transfer with the call transferred to him/her. For User C to transfer the transferred call, either user A or B need to first hang up, then user C can transfer call to user D.

- ✓ *If you need to transfer the call to the 4th party, you need to write settings in the Dial Plan of the bundled SIP server to redirect the call from User A to C to Brekeke PBX.*

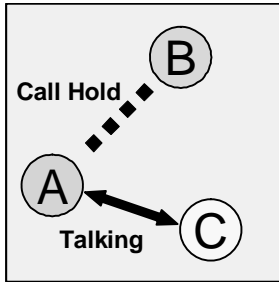


Figure 21

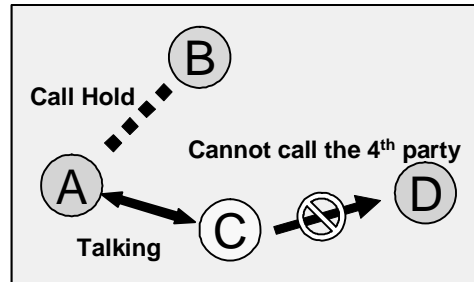


Figure 22

10. Auto Attendant

The Auto Attendant is a feature that interactively traffics calls without the intervention of a human operator.

10.1. Setting Up Auto Attendant

Set up for Auto Attendant is done by the System Administrator through the Brekeke PBX Admintool.

10.1.1. Accessing Auto Attendant

Extension: 001

Forwarding destinations	%ivr001
IVR	Auto Attendant

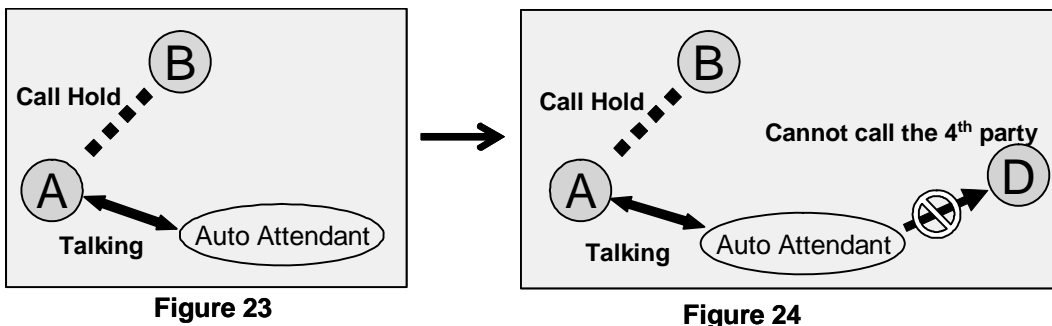
Add % and IVR prefix to the extension 001 and set it to [Forwarding destinations]. By adding % in front, user settings of the extension 001 will be applied for the call to this auto attendant. (Please refer to the section 7.1.5. "Example 5: Ignore Destination Number's Call Forward Settings-Forward Call Only to the Specified Extension Here").

- ✓ %ivr001 will be automatically set when you select the value other than "None" for the [IVR] setting.
- ✓ By adding % and IVR prefix to the extension number, you can play user 001's music on hold file for auto attendant.

10.2. Cautions for Auto Attendant

10.2.1. Call Transfer to Auto Attendant

When a call is transferred (Attended Transfer) to the Auto Attendant (Figure 23), Auto Attendant cannot transfer the call (Figure 24). Refer to section 9.6.1 "Extending Call Transfer".



However, Call Transfer to the 4th party is possible when the Call Transfer is initiated from another Auto Attendant. Since Auto Attendant operates automatically and the first Call Transfer is done instantly, 2nd Auto Attendant can transfer the call to the User D. (Figures 25 and 26)

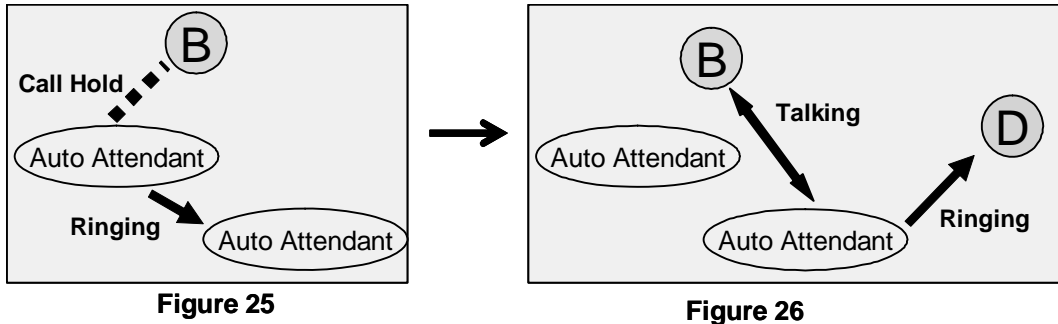


Figure 25

Figure 26

10.2.2. Call Transfer from Auto Attendant to the Non-Brekeke PBX User

Even when **[Transfer to unregistered users]** is disabled under **[Auto Attendant Settings]**, Call Transfers from Auto Attendant can be made if the non-Brekeke PBX User's number or SIP URI are set at **[Default operator]** field or **[Speed dial]** field.

10.3. Speed Dial Setting Example for Auto Attendant

10.3.1. Simple Speed Dial settings

Extension: 001

Auto Attendant Settings

Default operator	100
Speed dial*	6=100, 7=101

Extension: 100

Forwarding destinations*	100, 200
---------------------------------	----------

It is handy to set up **[Speed dial]** for using the Auto Attendant feature.

Your Auto Attendant guidance might say: *"If you know your party's extension, please enter it now. For sales, press 6. For support, press 7."*

If 6 or 7 are pressed, the call will be immediately transferred to the appropriate destinations. Extension 100 has defined multiple destinations for its **[Forwarding destinations]** setting as Ring Group setting. If there is no entry from the caller, the call will be transferred to the destination that is set in the **[Default operator]**.

✓ *With this setting, you will not be able to use the numbers that start with 6 or 7 for extension numbers.*

10.3.2. Speed Dial settings using Regular Expressions

Extension: 001

Auto Attendant Settings

Speed dial*	6636(.*)=\$1,(.*)=
Transfer to unregistered users	enable

Only when the dialed number is 6636 + <number>, the call will be transferred to <number>. Other numbers are not allowed. Because [Transfer to unregistered users] is set to enable, it is possible to transfer the call to an external user (such as PSTN number) but the caller need to know the first 4 digits (6636).

11. Voicemail

Each Brekeke PBX User can have a voicemail inbox.

11.1. Voicemail Inbox Number

The Brekeke PBX system treats a voicemail inbox number the same as a User ID or extension number.

11.2. Accessing Voicemail

Users can access their voicemail inbox by pressing the voicemail prefix number plus their extension number. For details, refer to Section 6.2 “Prefixes”.

Examples of accessing user extension 001’s voicemail inbox:

- ◆ Dialing 07*001 accesses the user 001’s voicemail inbox directly to leave messages
- ◆ Dialing 08*001 retrieves the user 001’s messages, change settings for voicemail inbox
- ◆ Pressing “*” (Star) while listening to the voicemail guidance same as dialing 08*001

Each user’s voicemail inbox is the default destination of **[Call forward (No answer)]** and **[Call forward (Busy)]**. For details about **[Forwarding destination (No answer/Busy)]**, refer to the section 7 “Call Forwarding”.

When a voicemail inbox receives a call, specified guidance is given to the caller. You can choose voicemail guidance from two personalized greetings and the default greeting.

11.3. Greeting Messages

11.3.1. Recording / Uploading Greeting Messages

Users can record their personalized greeting messages through **[Review / Option]** (Refer to “Voicemail Navigation Map”) or upload them through Brekeke PBX Admintool.

11.3.2. Selecting a Greeting Message

Users can have two personalized greeting messages, “Personal Greeting” and “Alternative Greeting.” Common use of the “Alternative Greeting” is for special occasions such as long holidays, sick days, and business trips.

You can also select a greeting message by calling the voicemail inbox and accessing **[Review / Option]**, or through the Brekeke PBX Admintool.

11.4. Managing Messages

The maximum length of the voicemail message that callers can leave is set in the Administrator's [Options] menu. The maximum number of voicemail messages that users can save in their inboxes is also set in the [Options] menu. Users can also set to forward the voicemail messages to specific extensions or destinations. For details, please refer to the section 11.8 "Voicemail Messages: Forwarding and Email Notification."

11.5. Retrieving Voicemail Messages

Retrieving voicemail can be done in several ways.

- ◆ You can download the sound file (WAV file) from the [Voice Mail] menu in Brekeke PBX Admintool. Once you download the file, you can listen to the file.
- ◆ By calling your voicemail inbox and accessing [Review / Option] menu
- ◆ By using Email notification setting, and opening sound file (WAV file) attached to the email

11.6. Voicemail Message Status

Your voicemail has two possible states, New message or Saved message.

- ◆ New message: a message that you have not saved or erased including newly arrived messages
- ◆ Saved message: a message that you have chosen to save on the system

11.6.1. New Message Status

This status is for the newly arrived voicemail messages, including forwarded messages from other users. This message will remain in the inbox up to the number of days set in the **[Days to keep unsaved messages]** in the [Options] menu.

11.6.2. Saved Message Status

Users can save voicemail messages when they are in message review. Saved messages will **not** be deleted even after the days set in the **[Days to keep unsaved messages]**.

11.7. Email Notification

When a new message arrives at your voicemail inbox, an email notification can be sent to a specified email address.

11.7.1. Set Up Email Notification

Detailed settings of Email Notification, such as email address to receive the notification or message file can only be set at the Brekeke PBX Admintool. Activation and deactivation of the email notification features can be changed by calling your voicemail access number without accessing Brekeke PBX Admintool. In the Brekeke PBX Admintool, you can set up your email address to receive notifications and choose whether or not to receive voicemail messages as an attachment with the notification. For details, please refer to the section 11.8 “Voicemail Messages: Forwarding and Email notification”.

11.7.2. Send Email Notification of the Messages from Call Recording

Recorded messages are stored in the same location as voicemail messages. Normally, email notification will not be sent for recorded messages. To send email notification for the recorded messages, additional variable is needed to set at [Options] > [Advanced]. Note that restarting Brekeke PBX is necessary to make this change to take effect.

Variable for setting call recording email notification:

```
com.brekeke.pbx.Setting.CALL_RECORDING_EMAIL_NOTIFICATION=true
```

11.8. Voicemail Messages: Forwarding and Email Notification

11.8.1. Forwarding Voicemail Message

If the extension user has set up voicemail message forwarding at **[Users > Edit] > [Voicemail settings] > [Message forwarding*]**, all voicemail messages will be forwarded to the specified destinations. You can assign multiple destinations to receive the voicemail messages. If you want to leave a copy of voicemail message as well as forwarding to destinations, you will need to specify your voicemail inbox number in the field as well. You can also forward voicemail messages to a group of destinations by using the forwarding settings.

11.8.2. Email Notifications

You may choose to receive email notifications when new voicemail messages arrive in your inbox. You can receive email notifications for messages that have been forwarded to other users as well.

Example:

- ◆ User 001 sets messages to forward to User 002. Only User 002 will receive voicemail message that came in to User 001
- ◆ User 001 enters both email addresses of User 001 and 002. Both User 001 and 002 will receive email notifications when User 001 receives a voicemail message

11.8.3. Max Messages

Brekeke PBX System Administrators can set the limit of the number of messages that users can have in their voicemail inboxes. **[Options] > [Media Server system setting] > [Max stored messages]**. When the number of voicemail messages in the inbox reaches the maximum number, callers cannot leave messages in your voicemail inbox. Callers will hear the system default message, "The message inbox is full."

A message forwarded to a user who has reached the maximum number of messages in their inbox will not be accepted. If you are forwarding your voicemail messages to multiple destinations, the message will also be forwarded to inboxes that have not reached the maximum number of messages.

11.9. Voicemail Inbox Access Limitation

Each voicemail inbox will only allow a single user access at a time. When you are accessing voicemail inboxes using a phone, you cannot modify your settings through the Brekeke PBX Admintool. When other is using your voicemail inbox, your voicemail will be locked to other accesses.

11.10. Creating / Deleting Voicemail Inboxes

This feature is only available to the system administrators. Creating or deleting a voicemail inbox is the same as creating or deleting a Brekeke PBX User extension. You may create/delete voicemail inboxes as follows:

Using the phone to create / delete voicemail inboxes:

- 1) Create a setup extension (such as, 1000) by choosing **[Users > Edit] > [IVR] > [Setup]**.
- 2) Call the setup extension number (1000) and follow the guidance to create/delete the voicemail inboxes (For details, please refer to "Voicemail Navigation Map.")

12. Automatic Route Selection (ARS)

Most of the call management rules for outgoing and incoming calls through outside lines are set at Automatic Route Selection.

Common outside lines that the PBX connects to:

- ◆ **Third Party SIP Servers**

Third Party SIP Server is server software which is not the bundled SIP server that comes with Brekeke PBX. Common third party SIP servers are those that are located at VoIP service providers, or other office locations.

- ◆ **Gateways**

The gateways referred to here are SIP PSTN Gateways that connect digital and analog PSTN lines or software gateways that convert between different protocols.

Defining rules enables ARS to make optimal use of outside lines, and your gateways. In the **[ARS]** menu, you can define extension numbers that will receive incoming calls, and which lines to use for outgoing calls.

Here are some examples of uses for ARS feature:

- ◆ **Select the least expensive calling options**

For example, you can send all international calls through a specific call long distance carrier, or to connect to the branch office located in Taipei. With ARS feature, the least expensive calling option can be utilized.

- ◆ **Effective port use of PSTN Gateways**

When there are multiple PSTN Gateways installed in the office, some ports may be used more frequently than others. The ARS feature can be used to select the most effective PSTN Gateway port automatically.

- ◆ **Effective Use of Outside Lines**

Similar to the above example, when you subscribe to multiple VoIP service providers, some phone numbers may be used more frequently than others. The ARS feature can be used to select the most effective service provider automatically.

- ◆ **Network Traffic Management and Load Balancing**

When many sessions are in progress and server load is too high or network bandwidth is insufficient, The ARS feature can be used to direct some calls through PSTN Gateways for example.

◆ **Failover (Pro Edition only)**

With ARS failover, Brekeke PBX searches for an alternate route if the specified route is unavailable or has failed, and makes outgoing calls via the alternative route. For example, if you have multiple ITSPs (Internet Telephony Service Providers) or/and PSTN (Public Switched Telephone Network) gateways, Brekeke PBX allows for backup connections if the VoIP (Voice over Internet Protocol) or internet connection is lost. In this situation, Brekeke PBX would automatically switch to an analog route through the PSTN gateway. The route can be re-activated manually from the Brekeke PBX Admintool or automatically by preset settings.

12.1. Route Search Process

Two sessions are used for one call in Brekeke PBX. Please refer to the section 5 “Call Process Flow” for the details of session processing during calls.

When A calls B, the SIP session which is initiated by A will be answered by Brekeke PBX. Brekeke PBX will process ARS Pattern-IN for the incoming session. Also Brekeke PBX initiates a session with B. Brekeke PBX will process ARS Pattern-OUT for the outgoing session.

Route Search is done by searching for Matching Patterns that apply to a call. The Route Search starts from the highest priority (lowest number) as defined by the ARS rules. If multiple patterns share a priority level, a pattern is selected randomly. When a matching pattern is found, the process will check if the Max Sessions value has been reached. If the pattern has sessions available, the process will select the pattern and convert the From and To header fields according to the settings defined in [Deploy Patterns].

For incoming sessions, Brekeke PBX will check if any ARS route matches and the session will be routed to an appropriate dial-in number. You also need to take care to ensure that calls between extension users do not match ARS rules, because those calls need not be routed through your gateway and third party SIP provider.

For outgoing calls, it is necessary to set priorities for effective routing depending on the usage of lines.

12.2. Matching Patterns

Using regular expressions enter conditions for To or From headers, etc. for the matching process. If the fields are left blank, all calls will be considered to match the condition. The fields can be set in the same way as [Matching Patterns] fields in the Dial Plan. If the fields are left blank, any call will match the rule.

Using [User] field of ARS Pattern-OUT setting, it is possible to restrict users for using the ARS pattern. "User" of the call for the ARS rule will be determined as follows:

- 1) If the destination number is Brekeke PBX user, the destination number is considered as "User" of the call.
- 2) If 1 is not applicable and also if the calls is forwarded through some Brekeke PBX user's [Forwarding destinations], [Forwarding destination (No answer)] or [Forwarding destination (Busy)] setting, the Brekeke PBX user will be considered as "User" of the call.
- 3) If 2 is not applicable and also if the caller is Brekeke PBX user, the caller will be considered as "User" of the call.
- 4) If none of the above is applicable, no user will considered as the "User" of the call.

[Plugin], [Param], [Return] fields are used only when you want to extend the matching conditions using a JAVA plug-in. Please refer to "ARS Plug-in Developer's Guide".

12.3. Deploy Patterns

This is similar to [Deploy Patterns] fields in Dial Plan; however, there are several important differences:

- ◆ Numbers follow the "\$" sign in [Deploy Patterns] in ARS rules. (Use "%" in Dial Plan)
- ◆ Conversion of From header that matches in [Matching Patterns] is done as described in [Deploy Patterns] for From header. Conversion of To header is done as defined in [Deploy Patterns] for To header, DTMF and Target field (In Dial Plan rules, multiple lines can be defined in [Matching Patterns] fields and [Deploy Patterns] fields and it is process at the same time).
- ◆ By using &f, &t, &u before number, it is possible to use a value of each [Matching Patterns] field in [Deploy Patterns] fields. &f, &t, &u correspond to From, To, and User.
- ◆ If From or To field is blank, no conversion will be performed. If DTMF field is blank, no DTMF will be sent. If Target field is blank, no target IP address will be specified (The call will be routed to the domain specified in To field in most cases).

12.4. ARS Templates and Editing Variables

It is possible to create a template in ARS Settings for multiple ARS rules that have same configuration patterns. In [Running Status], real rules (not template) will be displayed. In the template (excluding fields, [Route name], [Descriptions], and [Parameters]), you can use the variables &v1, &v2, ..., \$v9. You can specify real values in [Edit variables].

13. Third Party Call Control

Using this feature, you can connect multiple SIP UAs through an external interface. With this, you can create systems like “Click-to-Call” and “Autodialer”.

Using HTTP-GET (or POST) and Web Service interface, you can access Brekeke PBX externally. Here we will explain the method using HTTP-GET.

- 1) Set the allowed IP patterns at Brekeke PBX Admintool >[Options] > [PBX system settings] > [Valid client IP Pattern] using Regular Expressions.
- 2) Access the URL (format as below) using HTTP protocol from the PC with the IP addresses set at [Valid client IP Pattern].

URL format

http://<server-ip-address>:<server-port>/pbx/3pcc?to=<callee-number>&from=<caller-number>&user=<caller-user>&type=<type>

<server-ip-address>	IP address of server where Brekeke PBX is installed on
<server-port>	The same port number you use accessing Brekeke PBX Admintool
<callee-number>	Callee's number, or its SIP URI. You can specify multiple numbers by dividing number using comma.
<caller-number>	Caller's number or its SIP URI.
<caller-user>	Brekeke PBX User which is used to call out (caller). You can omit this if this is same with “caller-number”.
<type>	Specify “1” when you would like to call destinations simultaneously. Specify “2” when you want to connect Caller before connecting Callee.

Response

Brekeke PBX will return the response: “succeeded: ID number”. This ID will be the ID used at [Call Status] screen for the session.